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## SECTION 6

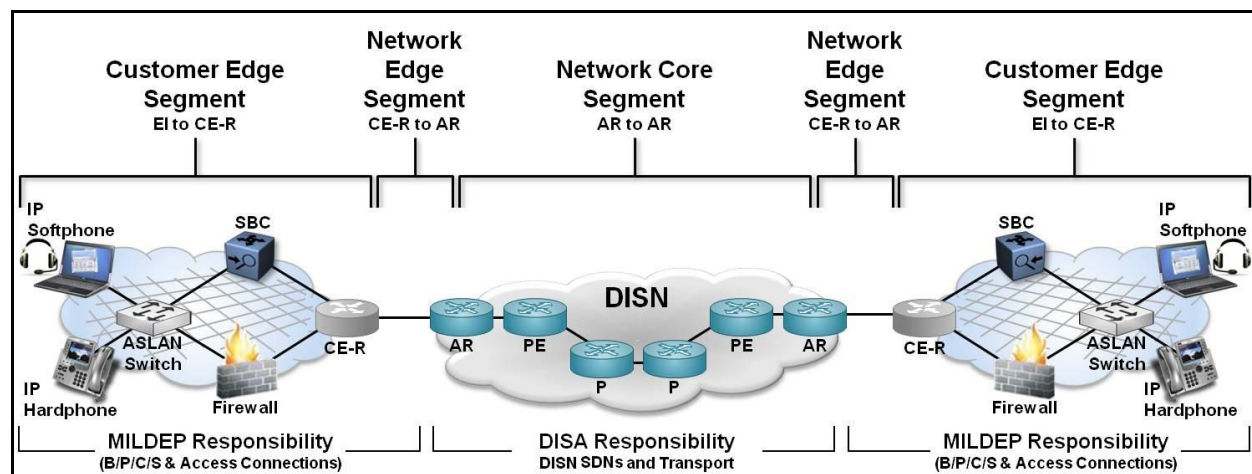
### NETWORK INFRASTRUCTURE END-TO-END PERFORMANCE

This section contains end-to-end (E2E) performance design guidelines for Unified Capabilities (UC) network infrastructures. The focus of this section is network-level design recommendations to attain optimal quality of service and service level objectives necessary for UC, while requirements for products used in the network infrastructure are defined in the Unified Capabilities Requirements (UCR) as follows:

1. The requirements for Local Area Network (LAN)/network edge products (i.e., LAN Core, Distribution, and Access switches; Customer Edge [CE] Routers [CE-Rs]) are provided in Section 7, Network Edge Infrastructure.
2. The requirements for the Network Infrastructure products (i.e., Defense Information Systems Network [DISN] Router, DISN Switch, and DISN Access Elements) are provided in Section 10, Network Infrastructure Products.
3. The Differentiated Services Code Point (DSCP) Plan, per-hop behavior (PHB), and traffic conditioning requirements for routers used in the UC network infrastructure are defined in UCR 2013, Section 6, Network Infrastructure End-to-End Performance.

#### 6.1 NETWORK SEGMENTS AND MEASUREMENT POINTS

The E2E network infrastructure consists of three network segments: CE, Network Edge, and Core. These are illustrated in [Figure 6.1-1](#), Measurement Points for Network Segments.



**Figure 6.1-1. Measurement Points for Network Segments**

This document does not intend to provide guidance on the many network deployment and implementation techniques, such as queuing strategies, traffic shaping, routing topology, or redundancy designs, that can be used freely by the network administrators to meet the required service objectives and their own internal needs. Its primary purpose is to provide a high-level design recommendation that, coupled with the performance metrics found in the UCR, can be

used to implement an optimal Internet protocol (IP)-based voice, video, and data architecture. These performance objectives are based on commercial best practices for latency, packet loss, jitter, and availability.

[Figure 6.1-1](#), Measurement Points for Network Segments, illustrates the components of the E2E network where measurements should be made to ascertain compliance with the service level objectives.

For the purpose of this document, the DISN Service Delivery Nodes (SDNs) are assumed to be bandwidth rich and robust. In addition, since the Assured Services LANs (ASLANs) are required to be implemented as nonblocking network entities for voice and video traffic, it is assumed that there is no bandwidth limitation in those segments as well. The access circuit, which may include a satellite communications (SATCOM) link, is the only potential bandwidth bottleneck. Therefore, the network design includes the use of Assured Services Admission Control (ASAC) to prevent session overload and subsequent voice and video performance degradation egressing from the CE.

The DISN Core provides high availability (99.96 percent or greater) using dual-homed and multi-homed access circuits with Multiprotocol Label Switching (MPLS) Fast Failure Recovery (FFR).

## 6.2 UC ENGINEERING NETWORK CONSIDERATIONS

The primary performance driver for UC is voice. Voice quality is calculated E2E from handset to handset. For voice applications, the measurement model for the End Instrument (EI) is the E-Model as described in the Telecommunications Industry Association (TIA)/TSB-116 A, which is based on the International Telecommunications Union – Telecommunication Standardization Sector (ITU-T) Recommendation G.107. The E-Model uses an R-Factor rating, which correlates to the Mean Opinion Score (MOS) rating specified by ITU-T recommendation P.800. The detailed EI voice quality calculation recommendations are found in UCR 2013, Section 2.20, Accounting Management.

This section specifies network infrastructure-related performance recommendations and takes into account all elements of the network to ensure that handset-to-handset recommendations are achievable. The following assumptions were made in determining performance recommendations necessary to achieve acceptable service:

- IPv4 or IPv6.
- Wireline Fixed Network (A=0).
- G.711 codec with 20 ms samples (Ie=0).
- IPv4 Bearer packet size = 254 bytes, calculated as the sum of the following:
  - Ethernet header – 22 bytes (Including optional VLAN Tag header)
  - IPv4 Packet header – 20 bytes

- 
- UDP header – 8 bytes
  - Real-Time Transport Protocol (RTP) header – 16 bytes (RFC 3550; section 5.1)
    - 2 bytes – Version, Padding, Extension, contributing source (CSRC) count, marker and payload type
    - 2 bytes – Sequence number
    - 4 bytes - Timestamp
    - 4 bytes - Synchronization source (SSRC) identifier
    - 4 bytes - Contributing source (CSRC) identifiers
  - Secure Real-Time Transport Protocol (SRTP) authentication headers– 12 bytes (RFC 3711; section 3.1)
    - 4 bytes – Optional RTP Extension
    - 4 bytes – Optional Master Key Identifier (MKI)
    - 4 bytes – Recommended Authentication Tag
  - G.711 - 160 bytes
    - $64000 \text{ bps (Bit Rate)} * .020 \text{ seconds (Sampling Rate)} = 1280 \text{ bits (160 bytes)}$
  - Ethernet Frame Check Sequence – 4 bytes
  - Ethernet Interframe Gap – 12 bytes
  - IPv6 Bearer packet size = 274 bytes, Calculated as the sum of the following:
    - Ethernet header – 22 bytes (Including optional VLAN Tag header)
    - IPv6 Packet header – 40 bytes
    - UDP header – 8 bytes
    - Real-Time Transport Protocol (RTP) header – 16 bytes (RFC 3550; section 5.1)
      - 2 bytes – Version, Padding, Extension, contributing source (CSRC) count, marker and payload type
      - 2 bytes – Sequence number
      - 4 bytes - Timestamp
      - 4 bytes - Synchronization source (SSRC) identifier
      - 4 bytes - Contributing source (CSRC) identifiers
    - Secure Real-Time Transport Protocol (SRTP) authentication headers– 12 bytes (RFC 3711; section 3.1)
      - 4 bytes – Optional RTP Extension
      - 4 bytes – Optional Master Key Identifier (MKI)
      - 4 bytes – Recommended Authentication Tag

- G.711 - 160 bytes
  - $64000 \text{ bps (Bit Rate)} * .020 \text{ seconds (Sampling Rate)} = 1280 \text{ bits (160 bytes)}$
- Ethernet Frame Check Sequence – 4 bytes
- Ethernet Interframe Gap – 12 bytes
- Weighted Terminal Coupling Loss (TCLw) = 52 dB (in accordance with [IAW] American National Standards Institute [ANSI]/ Telecommunications Industry Association [TIA]-810-B).
- Latency because of EI (voice to IP and IP to voice) is 50 ms:
  - Latency because of de-jitter buffer in an EI is 20 ms.
  - Includes Packet Loss Concealment delays.

### 6.2.1 Voice Codec Compression

The preferred codec used for E2E Fixed-to-Fixed (F-F) voice sessions is the G.711 Pulse-Code Modulation (PCM) (Uncompressed) with 20 ms samples. Other codecs are allowed, and a minimum list of codecs that must be supported by all EIs is found in Section 2, Session Control Products.

## 6.3 ASSURED UC LATENCY DESIGN CONSIDERATIONS

The one-way latency metric is reported as the arithmetic mean of several (specified) single measurements over a 5-minute period. Corrupt and lost packets are excluded from the calculation. The metric is reported to 1 ms accuracy, rounded up, with a minimum value of 1 ms.

### 6.3.1 Assured UC Router Serialization/Packet Switching Latency

UCR 2013, Section 6, Network Infrastructure End-to-End Performance, specifies that all routers must be capable of receiving, processing, and transmitting a voice packet within 2 ms or less in addition to the serialization delay for voice packets as measured from the input interface to output interface under congested conditions, to include all internal functions. For example, the serialization delay of a 100 Base-T interface is 0.017 ms, which would allow for voice latency from input to Ethernet output under congested conditions of 2.017 ms.

NOTE: Internal functions do not include Domain Name Service (DNS) lookups and other external actions or processes.

### 6.3.2 Assured UC End-To-End Latency

The E2E network infrastructure supporting UC must ensure that the one-way E2E latency (handset to handset) for F-F locations does not exceed 220 ms for UC sessions as averaged over any 5-minute period. [Figure 6.3-1](#), F-F E2E Latency, illustrates the measurement points for calculating the F-F E2E latency.

NOTE 1: The 220 ms in this context is taken from the E-Model ITU-T G.107

Recommendation, that states that a 220ms one-way delay is equal to a MOS of 4.0.

NOTE 2: The recommendation for 220 ms is due to the limits of talk over. This latency may not be feasible for all scenarios (i.e., Southwest Asia [SWA]), but the recommendation as stated is necessary to avoid talk over for the scenarios that are feasible.

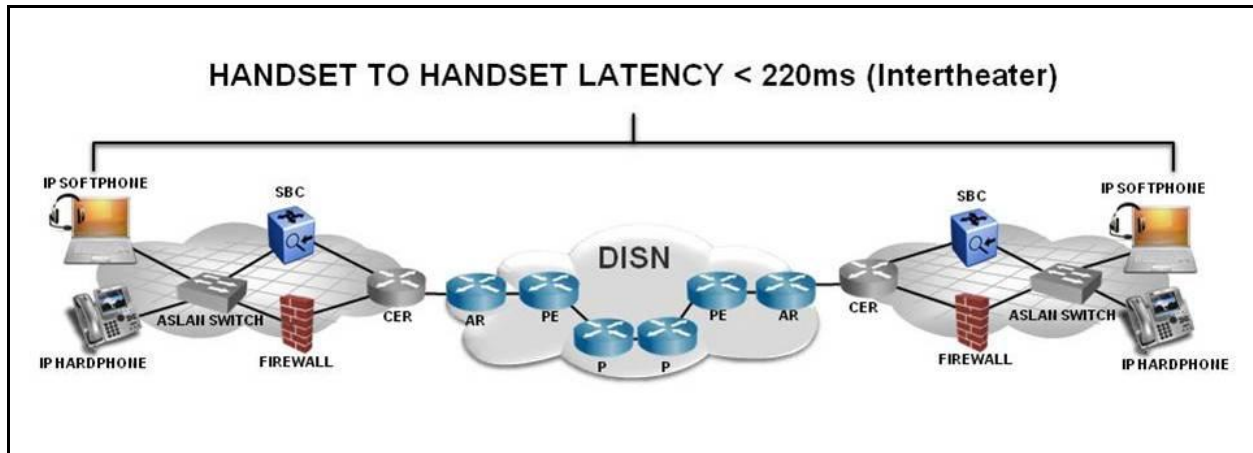
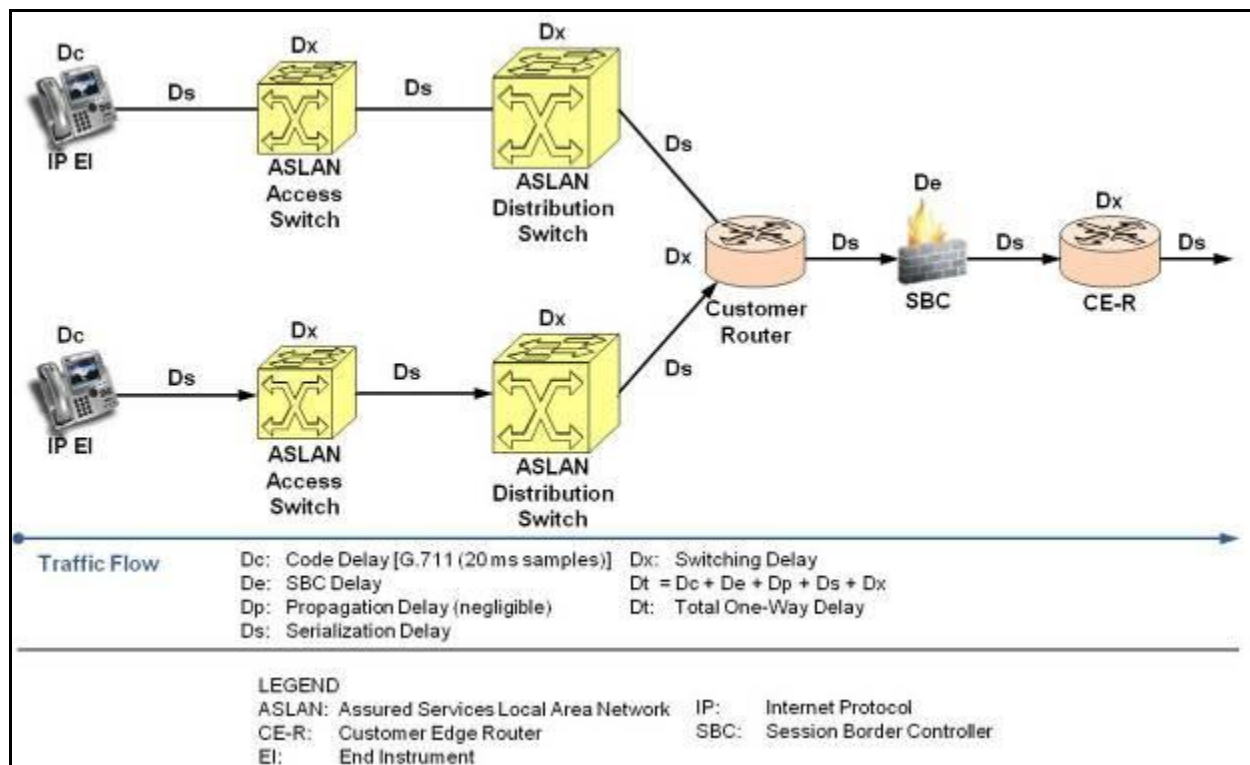


Figure 6.3-1. F-F E2E Latency

### 6.3.3 Assured UC CE Segment Latency

The CE Segment supporting UC must ensure that the one-way latency from the IP handset to the egress interface of the CE-R within the CE Segment is less than or equal to 35 ms (or less than or equal to 44 ms if the CE-R is collocated with an Aggregation Router [AR]) for UC sessions as averaged over any 5-minute period. [Figure 6.3-2](#), CE Segment Outbound Latency, illustrates the delays associated with calculating the CE Segment outbound latency. The measurements must include the latency associated with the CE-R packet switching.



**Figure 6.3-2. CE Segment Outbound Latency**

The CE Segment supporting UC must ensure that the one-way latency from the ingress interface of the CE-R to the IP handset within the CE Segment is less than or equal to 35 ms (or less than or equal to 44 ms if the CE-R is collocated with an AR) for UC sessions as averaged over any 5-minute or period. [Figure 6.3-3](#), CE Segment Inbound Latency, illustrates the delays associated with calculating the CE Segment inbound latency. The measurements must include the latency associated with the CE-R packet switching.

NOTE: This assumes that the latency associated with the de-jitter buffer is 20 ms.

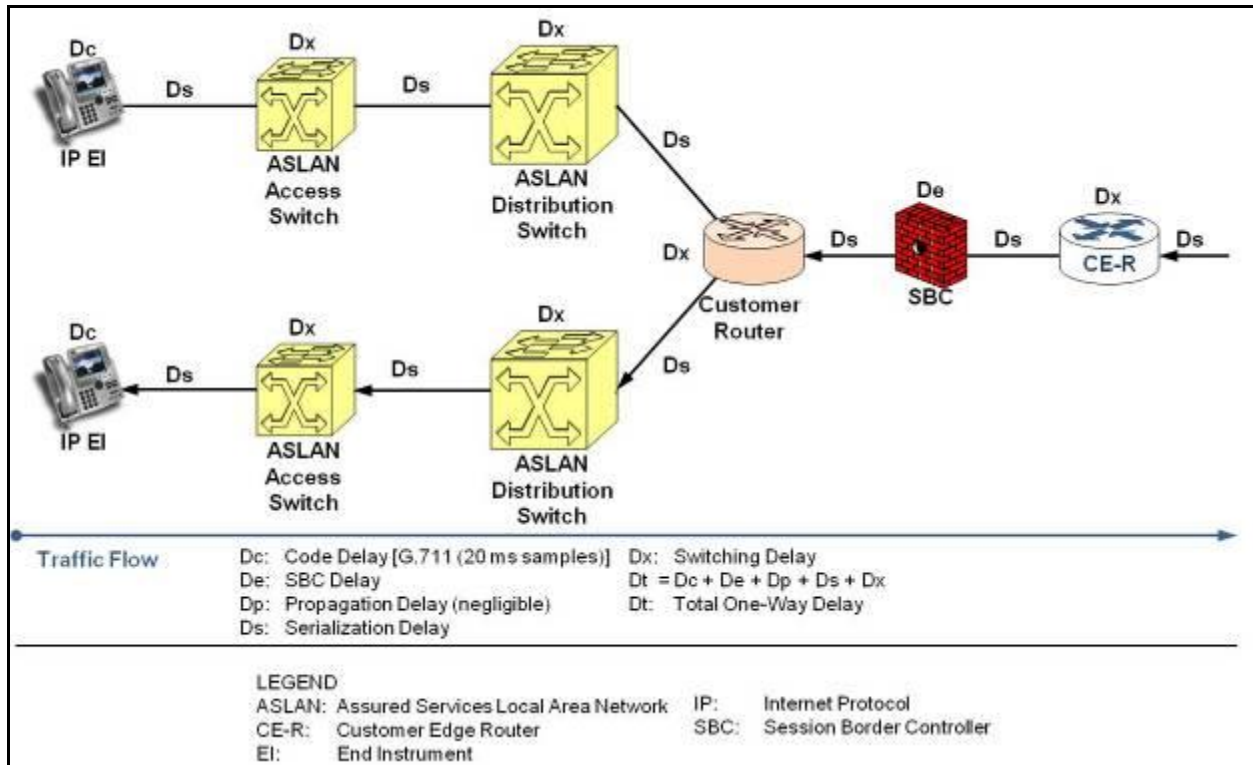
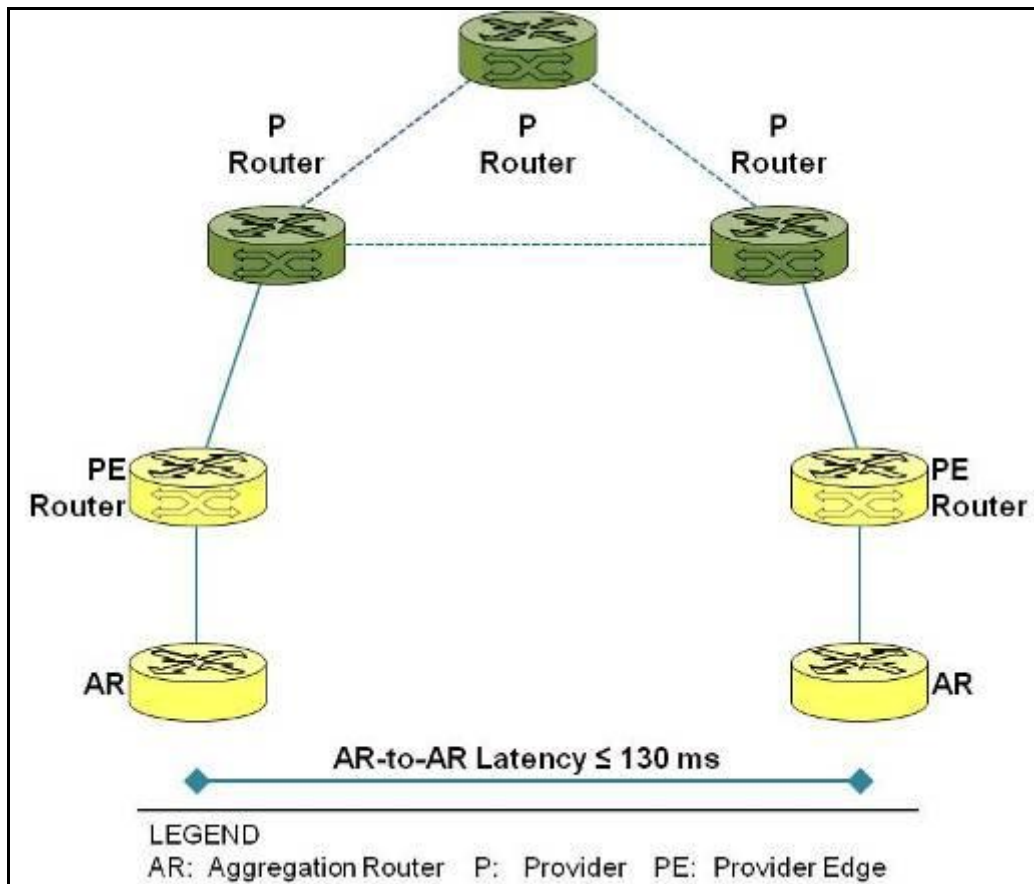


Figure 6.3-3. CE Segment Inbound Latency

### 6.3.4 Assured UC AR-to-AR Latency

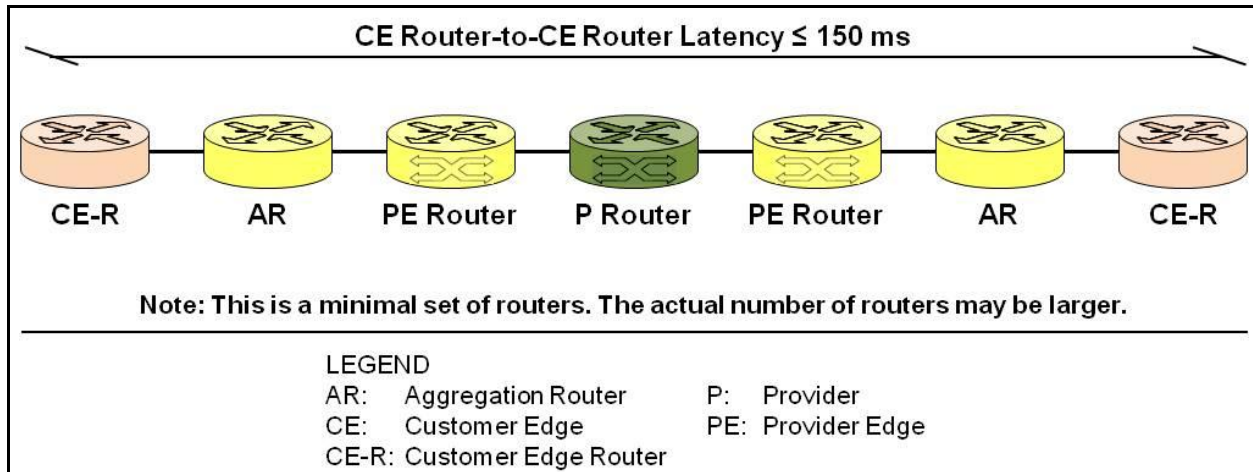
The network infrastructure supporting UC must ensure that the one-way latency measured from the ingress interface of an AR to the egress interface of another AR across the DISN Wide Area Network (WAN) for F-F nodes does not exceed 130 ms latency for UC sessions averaged over any 5-minute period. The measurement must take place between interfaces facing the CE-R to incorporate the packet switching delays through the network device. [Figure 6.3-4](#), F-F AR-to-AR Latency, illustrates the measurement points for calculating the F-F AR-to-AR latency.



**Figure 6.3-4. F-F AR-to-AR Latency**

### 6.3.5 Assured UC CE Router-to-CE Router Latency

The DISN Network Infrastructure supporting UC must ensure that the one-way latency measured from the ingress interface of a CE-R to the egress interface of another CE-R across the DISN Network Infrastructure for F-F nodes does not exceed 150 ms (or 132 ms if the CE-R is collocated with an AR) for UC sessions averaged over any 5-minute period. The measurement must take place between interfaces inclusive of the traffic flow through the CE-R to incorporate the packet switching delays through the network device. [Figure 6.3-5](#), F-F CE Router-to-CE Router Latency, illustrates the measurement points for calculating the F-F CE Router-to-CE Router latency.



**Figure 6.3-5. F-F CE Router-to-CE Router Latency**

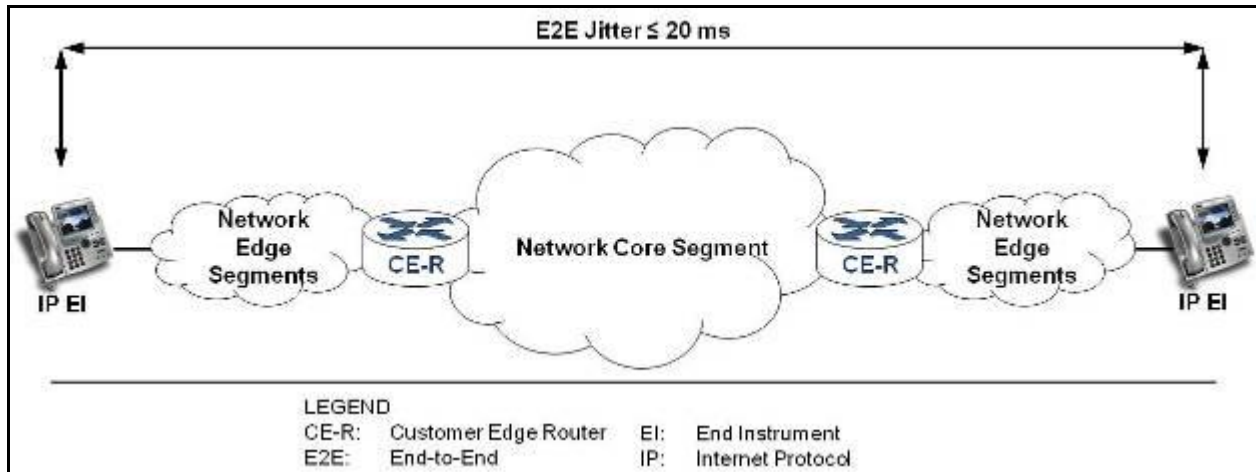
## 6.4 ASSURED UC JITTER

Jitter is defined in Appendix C, Definitions, Abbreviations and Acronyms, and References, and the term is used interchangeably with the term IP Packet Delay Variation (IPDV). The jitter numbers specified in this section are based on the minimum latency jitter model defined in ITU-T Recommendation Y.1540, November 2007. The one-way jitter is defined as the 99th percentile measurement of the distribution of singleton jitter (n) measurements over a 5-minute measurement interval.

### 6.4.1 Assured UC End-to-End Jitter

The E2E network infrastructure supporting UC must ensure that the E2E jitter (handset-to-handset) for F-F locations does not exceed 20 ms for UC sessions during any 5-minute period. [Figure 6.4-1](#), E2E F-F Jitter, illustrates the measurement points for calculating the F-F E2E network jitter.

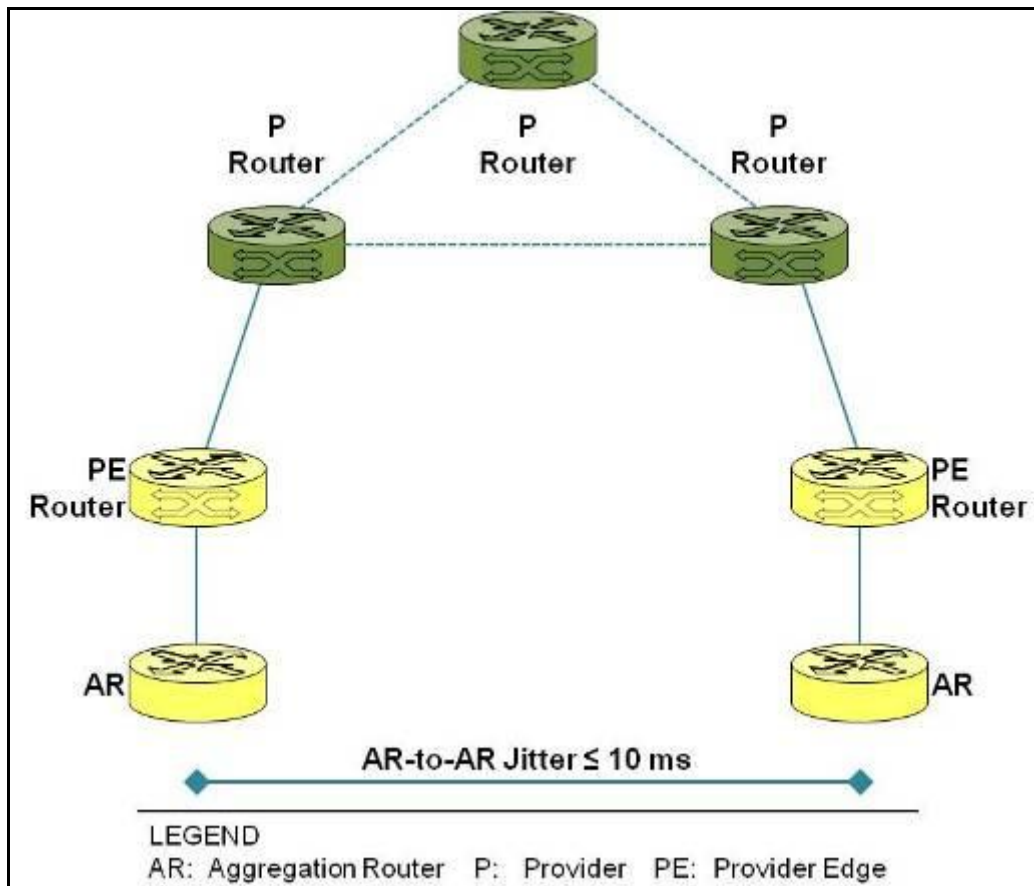
NOTE: Dynamic de-jitter buffers are allowed, but for these performance measurements are assumed to be 20 ms.



**Figure 6.4-1. E2E F-F Jitter**

## 6.4.2 Assured UC AR-to-AR Jitter

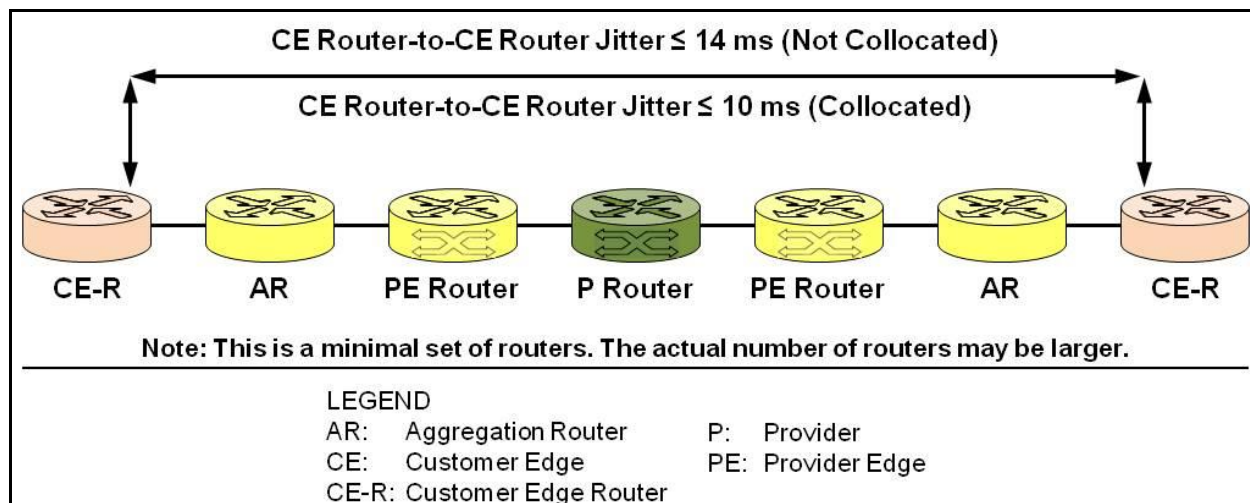
The network infrastructure supporting UC must ensure that the one-way jitter measured from the ingress interface of an AR to the egress interface of another AR across the DISN WAN for F-F nodes does not exceed 10 ms for UC sessions averaged over any 5-minute period. The measurement must take place between interfaces facing the CE-R to incorporate packet jitter delays through the network device. [Figure 6.4-2](#), F-F AR-to-AR Jitter, illustrates the measurement points for calculating the F-F AR-to-AR jitter.



**Figure 6.4-2. F-F AR-to-AR Jitter**

### 6.4.3 Assured UC CE Router-to-CE Router Jitter

The DISN Network Infrastructure supporting UC must ensure that the one-way jitter measured from the ingress interface of a CE-R to the egress interface of another CE-R across the DISN Network Infrastructure for F-F nodes does not exceed 14 ms (or 10 ms if the CE-R is collocated with an AR) for UC sessions averaged over any 5-minute period. The measurement must take place between interfaces inclusive of the traffic flow through the CE-R to incorporate packet jitter delays through the network device. [Figure 6.4-3](#), F-F CE Router-to-CE Router Network Infrastructure Jitter, illustrates the measurement points for calculating the F-F CE Router-to-CE Router network infrastructure jitter.



**Figure 6.4-3. F-F CE Router-to-CE Router Network Infrastructure Jitter**

#### 6.4.4 Assured UC CE Segment Jitter

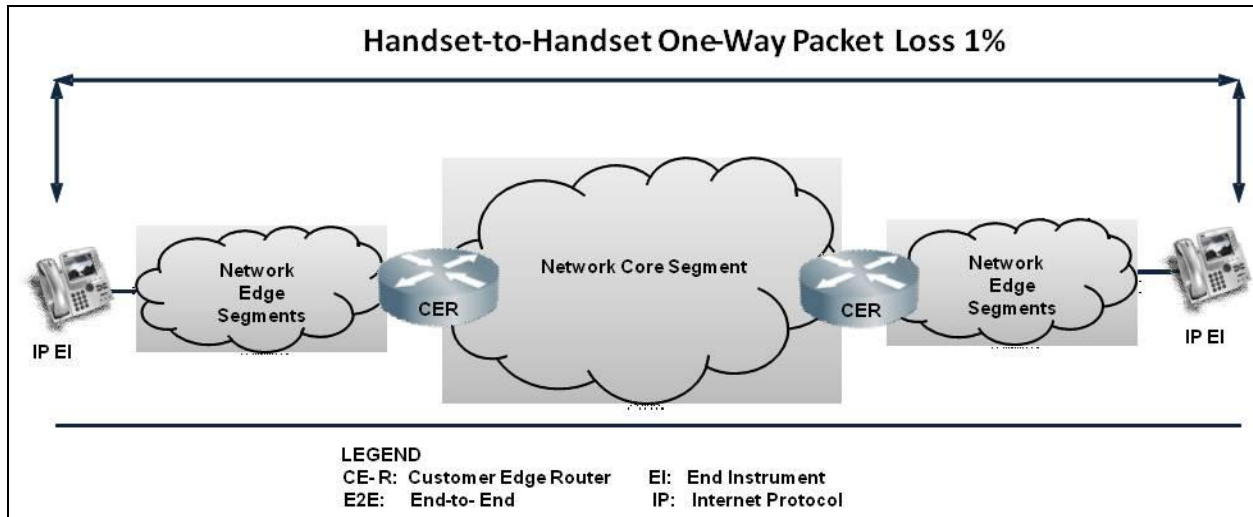
The CE Segment supporting UC must be configured not to exceed a one-way jitter of 3 ms (or 5 ms if the CE-R is collocated with an AR) measured between the handset and the egress interface of the CE-R for UC sessions during any 5-minute period.

### 6.5 ASSURED UC PACKET LOSS DESIGN CONSIDERATIONS

Packet loss is defined in Appendix C, Definitions, Abbreviations and Acronyms, and References, and the term is used interchangeably with the term IP Packet Loss Ratio (IPLR). A single instance of packet loss measurement is defined as a record of a packet sent by a sender reference point captured and compared at a destination reference point. The record is zero if the packet was received or one if the packet was not received. A packet is deemed to be lost if its one-way latency exceeds a time  $T_{\max}$ , where  $T_{\max}$  is equal to 3 seconds.

#### 6.5.1 Assured UC End-To-End Packet Loss

The E2E network infrastructure supporting UC must ensure that the E2E one-way IP packet loss, as measured from handset to handset, for current F-F deployed locations does not exceed 1.0 percent for UC sessions averaged over any 5-minute period. [Figure 6.5-1](#), E2E F-F Packet Loss, illustrates the measurement points for calculating the F-F E2E packet loss.

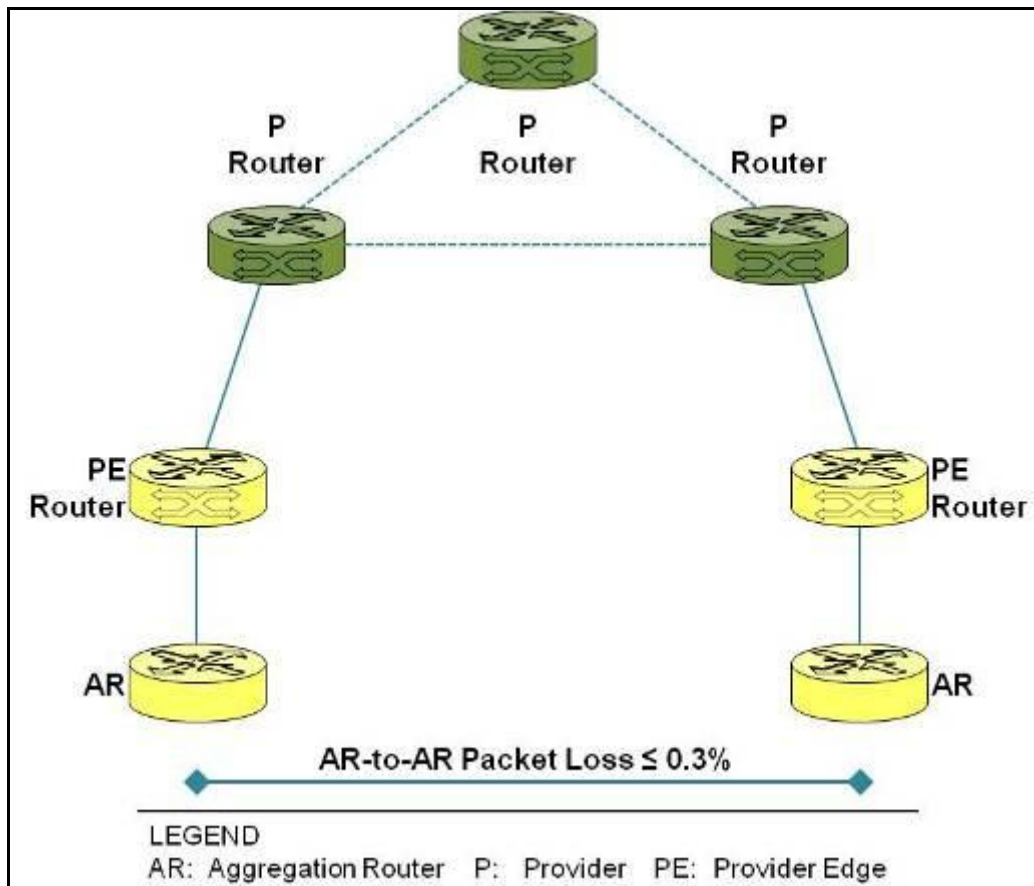


**Figure 6.5-1. E2E F-F Packet Loss**

On all new network deployments, the E2E network infrastructure supporting UC must be designed and engineered for a one-way E2E packet loss for F-F locations of 0 percent for UC sessions as averaged over any 5-minute period.

### 6.5.2 Assured UC AR-to-AR Packet Loss

The network infrastructure supporting UC must ensure that the one-way packet loss measured from the ingress interface of an AR to the egress interface of another AR across the DISN WAN for F-F nodes does not exceed 0.3 percent for UC sessions averaged over any 5-minute period. The measurement must take place between interfaces facing the CE-R to incorporate packet loss through the network device. [Figure 6.5-2](#), F-F AR-to-AR Packet Loss, illustrates the measurement points for calculating the F-F AR-to-AR packet loss.

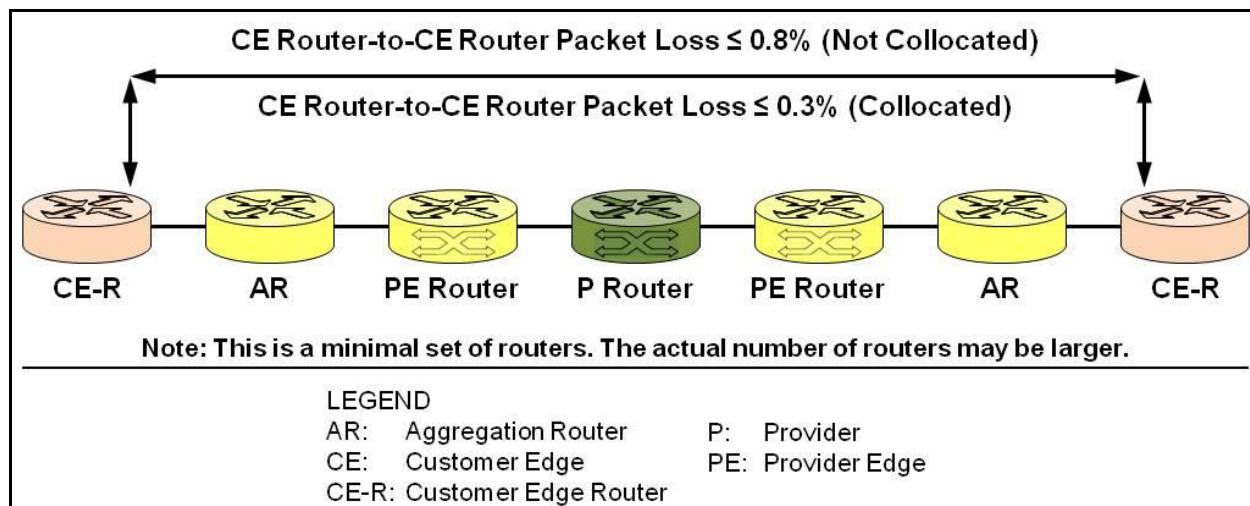


**Figure 6.5-2. F-F AR-to-AR One Way Packet Loss**

### 6.5.3 Assured UC CE Router-to-CE Router Packet Loss

The DISN Network Infrastructure supporting UC must ensure that one-way packet loss measured from the ingress interface of a CE-R to the egress interface of another CE-R across the DISN Network Infrastructure for F-F nodes does not exceed 0.8 percent (0.3 percent if the CE-R is collocated with an AR) for UC sessions averaged over any 5-minute period. The measurement must take place between interfaces inclusive of the traffic flow through the CE-R to incorporate packet loss through the network device.

[Figure 6.5-3](#), F-F CE Router-to-CE Router Network Infrastructure Packet Loss, illustrates the measurement points for calculating the F-F CE Router-to-CE Router packet loss.

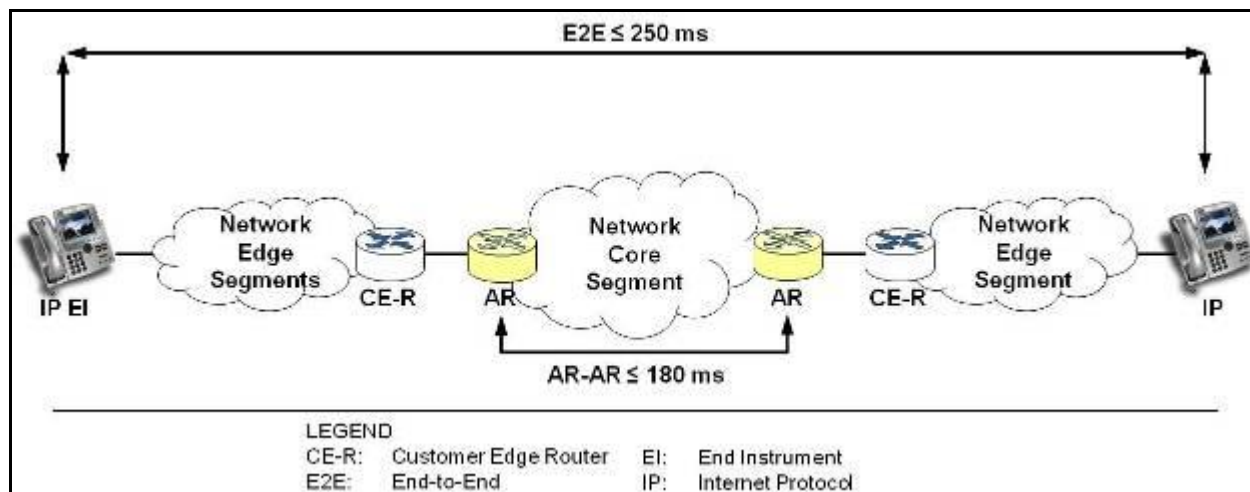


**Figure 6.5-3. F-F CE Router-to-CE Router Network Infrastructure Packet Loss**

NOTE: This assumes packet loss between a collocated CE-R and AR is 0.01 percent or less.

## 6.6 NON-ASSURED VOICE

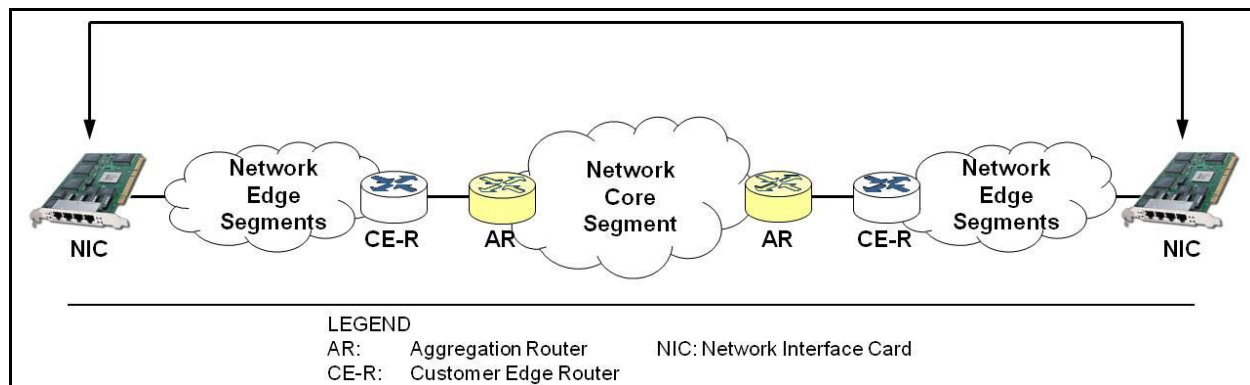
The performance objectives for assured Voice and non-assured Voice are the same except with respect to E2E and AR-to-AR latency. Unlike assured Voice, non-assured Voice does not support Command and Control (C2) users and does not require as stringent performance objectives as assured Voice. As a result, non-assured VoIP is engineered for a MOS of 3.8, which is consistent with the performance of commercial wireless voice services. In accordance with the G.107 MOS model, a MOS of 3.8 can be achieved with a packet loss of 1 percent using the G.711 Codec when the latency is less than 250 ms. Therefore, E2E and AR-to-AR performance objectives for non-assured Voice latency are no longer the same E2E and AR-to-AR performance objectives for assured Voice latency. The performance objectives for non-assured VoIP are listed in [Table 6.8-1](#), Granular Service Class Performance Objectives. Latency objectives for non-assured Voice are also illustrated in [Figure 6.6-1](#), Latency Objectives for Non-Assured Voice.



**Figure 6.6-1. Latency Objectives for Non-Assured Voice**

## 6.7 DATA APPLICATIONS

The performance of data applications will be measured at the same points as assured voice with the exception of E2E data performance, which will be measured from NIC to NIC (See [Figure 6.7-1](#)) instead of from IP Voice EI to IP Voice EI. The performance objectives for data applications are listed in [Table 6.8-1](#), Summary of Granular Service Class Performance Objectives.



**Figure 6.7-1. E2E Performance Measured from NIC to NIC**

## 6.8 SERVICE LEVEL SPECIFICATION

[Table 6.8-1](#) summarizes the Service Level Specification for each granular UC service class as defined in UCR 2013, Section 6, Network Infrastructure End-to-End Performance. This table defines one-way performance recommendations.

**Table 6.8-1. Service Level Class Specification**

<b>GRANULAR SERVICE CLASS</b>	<b>E2E LATENCY (MS)</b>	<b>AR-AR LATENCY (MS)</b>	<b>EI-CE-R LATENCY (MS)</b>	<b>E2E PACKET LOSS (%)</b>	<b>AR-AR PACKET LOSS (%)</b>	<b>EI-CE-R PACKET LOSS (%)</b>	<b>E2E JITTER (MS)</b>	<b>AR-AR JITTER (MS)</b>	<b>EI-CE-R JITTER (MS)</b>
Short Messaging	1000	900	50	0.5	0.4	0.05			
Assured Voice	220	150	35	1	0.8	0.05	20	14	3
Assured Multimedia Conferencing	220	150	35	1	0.8	0.05	20	14	3
Broadcast Video	1000	900	50	0.1	0.08	0.01			
Multimedia Streaming (includes Non-Assured Video)	250	180	35	1	0.8	0.05	20	14	3
Non-Assured Voice	250	180	35	1	0.8	0.05	20	14	3
Low Latency Data: IM/Chat, Presence	300	200	50	1	0.8	0.05			
High Throughput Data: Real-Time Data Backup, Web Hosting	300	200	50	1	0.8	0.05			
NOTE: Not All Aggregate Service Classes Have Performance Objectives (Best Effort, Signaling, Network Control, & Low Priority)									

## 6.9 SYSTEM-LEVEL QUALITY FACTORS

### 6.9.1 Handset-to-Handset Availability

The definition of availability, found in the Telcordia Technologies GR-512-CORE, Section 12, is the basis for the E2E UC network reliability. The following paragraphs outline the availability recommendations for the UC network.

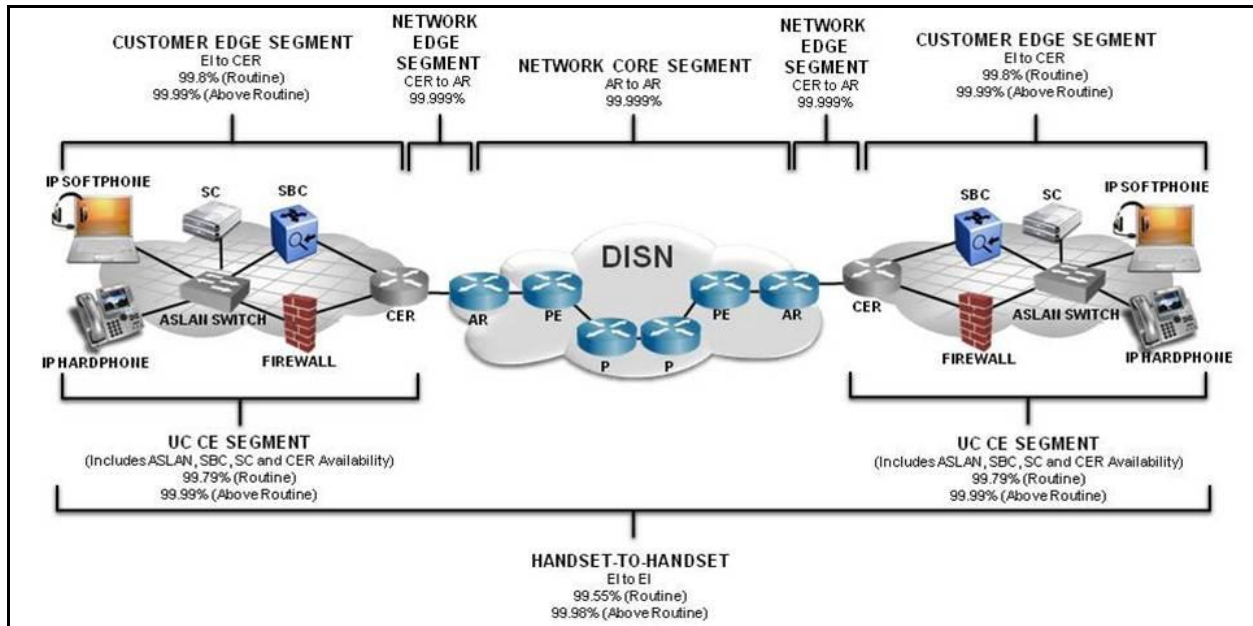
1. Handset-to-Handset UC availability takes into account network resiliency and the availability of the devices that enable UC communications. The availability for the handset-to-handset network infrastructure between F-F locations serving UC users inclusive of scheduled maintenance must be as follows:
  - a. ROUTINE (R) precedence of 99.55 percent or greater.
  - b. IMMEDIATE/PRIORITY (I/P) precedence of 99.98 percent or greater.
  - c. FLASH or FLASH OVERRIDE (F/FO) precedence of 99.98 percent or greater.

## 6.9.2 Network Segments Availability

1. The availability for the Network Core measured from AR to AR must be 99.999 percent or greater to include scheduled maintenance.
2. The availability for the network measured from CE-R to CE-R must be as follows:
  - a. R precedence of 99.951 percent or greater.
  - b. I/P, F, or F/O precedence of 99.987 percent or greater.
3. The availability for the CE Segment measured from EI to CE-R (ASLAN) must be as follows:
  - a. R precedence of 99.8 percent or greater.
  - b. I/P precedence of 99.997 percent or greater.
  - c. F or F/O precedence of 99.999 percent or greater.
4. The availability for the UC CE Segment measured from EI to CE-R (to include ASLAN, Session Border Controller [SBC], Session Controller [SC], and CE-R) must be as follows:
  - a. R precedence of 99.79 percent or greater.
  - b. I/P precedence of 99.99 percent or greater.
  - c. F or F/O precedence of 99.99 percent or greater.

NOTE: Availability calculations are based on best practices because there appears to be no standardized model for calculating IP network availability.

[Figure 6.9-1](#), F-F Network Infrastructure Availability, illustrates the measurement points for calculating the F-F network availability.



**Figure 6.9-1. F-F Network Infrastructure Availability**

### 6.9.3 Availability Design Considerations

1. The E2E network infrastructure supporting UC users with precedence above ROUTINE must have no single point of failure to include power sources and Network Management (NM).
2. If the network infrastructure supports users with precedence above ROUTINE, then the network infrastructure routers must provide five nines of availability (99.999 percent) to include scheduled maintenance.

NOTE: Network infrastructure router availability recommendations may be met using dual homing and other routing techniques.

3. The National Military Command Center (and Alternate), Combatant Commands (COCOMs), or Component headquarters must not be isolated longer than 30 minutes because of an outage in the Core Segment of the network.
4. In the event of an E2E network infrastructure component failure in a network supporting UC users with precedence above ROUTINE, all sessions that are active must not be disrupted (i.e., loss of existing connection requiring redialing) and a path through the network must be restored within 5 seconds.
5. If the Edge Segment is dual-homed or multi-homed, then the UC traffic must be engineered to only use one access connection at a time to prevent asymmetric routing.

NOTE: Data traffic should be engineered to use the alternate connection and serve as the UC traffic redundant link.

6. No segment of the E2E network infrastructure must use the same cost metric on dual-homed or multi-homed configurations forcing the same UC routing stream to be split into two distinct interfaces.

NOTE: Cost metric redundancy routing is a technique used to provide survivability by sending packets associated with a session across multiple paths through the network infrastructure. This technique often introduces unacceptable and hard to troubleshoot latency and jitter on real time services and applications.

7. All network infrastructure products supporting UC users with precedence above ROUTINE must have 8 hours of backup power.

NOTE: This recommendation does not address ASLAN backup power recommendations, which are addressed in Section 7, Network Edge Infrastructure.

8. If the Edge Segment supports users with precedence above ROUTINE, then the Edge Segment must be multi-homed, this means two separate access connections provisioned on physically diverse paths via two different ARs to two different service providers. Multi-homing is physically and logically diverse in accordance with the DISN subscription rates.
9. If the Edge Segment is dual-homed or multi-homed, and supports users with precedence above ROUTINE, then each connection must be traffic engineered to support 100 percent, which includes the 25 percent surge recommendation of the UC traffic load.
10. If the Edge Segment supports users with precedence above ROUTINE and the CE-R is collocated with an SDN containing a robust MSPP, then a separate access connection to another robust SDN must be used for redundancy.

#### **6.9.4 Reliability and Failover Considerations**

1. End-to-end network infrastructure products supporting UC users with precedence above ROUTINE must support a protocol that allows for dynamic rerouting of IP packets to eliminate any single points of failure in the network.
2. All network infrastructure products supporting UC users with precedence above ROUTINE used to meet the reliability recommendations must be capable of handling the entire session processing load in the event that its counterpart product fails.
3. All network infrastructure products supporting UC that implement MPLS must support a Fast Reroute (FRR) capability that restores routing paths following a local failure (i.e., a failure involving a single router or circuit) within 50 ms.
4. Network infrastructure routers must only enact switchovers based on a reduction in access network throughput or bandwidth with Network Management (NM) troubleshooting procedures, because the routers cannot determine where or what in the access IP connection is the cause of the reduction.

5. If the Edge Segment has at least two separate access connections and the CE-R detects an access connection failure, the CE-R must be configured to dynamically switch to the alternate or backup access connection.

NOTE: A failure may be detected via a physical link alarm (level 2), or via a loss of a dynamic routing protocol HELLO status message (level 3).

6. If the CE-R has at least two separate access connections (i.e., dual-homed or multi-homed) and detects an access connection failure, the CE-R must switch to the alternate or backup access connection using an automatic process and must not require operator actions.

NOTE: When the switchover occurs, UC sessions in progress may be lost, and new sessions may not be able to be established until the IP routing updates have taken place. This may take 10 seconds or more and is dependent on the routing protocol standard update interval.

7. If the Edge Segment has at least two access connections to provide redundancy, then the network administrators must implement a standard operating Procedure for switching UC traffic between access connections at least on a weekly basis to verify that the alternate circuit or path is working properly.

## 6.10 BANDWIDTH PROVISIONING CONSIDERATIONS

The recommended bandwidth per supported voice session on the Ethernet network infrastructure is 220 kbps (110 kbps each direction). This is based on the worst-case scenario assumption that uses a 274 bytes voice bearer packet as defined in [Section 6.2](#).

In addition, the E2E network infrastructure supporting UC must assume the use of G.711 (20 ms) for calculating bandwidth budgets within F-F networks even if compressed codecs are used. For example, if G.729 is used for an F-D UC session, then the budget for the fixed portion of the network should allocate 110 kbps to that session even though the session uses less bandwidth.

1. Access connections supporting UC must be engineered to support one WAN (trunk) voice session (110 kbps of IP bandwidth in each direction) for every four EIs within the Edge Segment

(NOTE: The 4:1 ratio does not include surge); or must be traffic engineered IAW the following approach:

- a. Determine the busy hour traffic load in Erlangs from current traffic pattern, matrix, or call volume using the following formula and use the Erlang B table to determine the number of connections/size of connection required to support the traffic load.

Busy Hour Offered Load = Total Call Time for the Busy Hour in Seconds/10  
(averaged over the 10 busiest hours of the year)

Busy Hour Erlang = Busy Hour Offered Load in Seconds/3600

- b. Calculate the Access Connection bandwidth recommendation based on the following assumptions:

Call Arrival Distribution	=	Poisson
Codec Type	=	G.711 (coding rate: 64000 bits/sec)
Frame Size	=	20 ms interval time (0.020 sec)
Samples/Packet	=	160 samples per packet
Frames/Packet	=	1
Frames/Second	=	50
Frame Size/Packet	=	160 bytes
Ethernet Interframe Gap	=	12 bytes
SRTP Authentication Tag	=	4 bytes
Frames/Erlang	=	50
Packets/Second/Erlang	=	50
Packet Size (for Ethernet)	=	274 bytes (assumes IPv6)
Access Bandwidth Formula	=	Busy Hour Erlang B * Packet Size * Packets/Second/Erlang B * 8 bits/byte

For example, if the Busy Hour Erlang B equals 25, then the access bandwidth should be  $25 * 274 * 50 * 8 = 2,740,000$  bits per second (bps) or 2.74 Mbps.

2. A Base/Post/Camp/Station (B/P/C/S) must not reduce the number of simultaneous Access Connection (trunk) subscriptions to the DISN when they migrate from Time Division Multiplexing (TDM) to IP unless traffic engineering is completed IAW the preceding recommendation.  
  
NOTE: For instance, if the existing B/P/C/S subscribed for 100 simultaneous DS0s to the DISN with their TDM infrastructure, but the engineered IP solution only requires 90 multiplied by 110 kbps of bandwidth, then the B/P/C/S design must support 100 multiplied by 110 kbps of bandwidth to meet this recommendation.
3. The E2E network infrastructure design must provide, at a minimum, a 25 percent increase in network capacity (i.e., throughput and number of sessions) above the current employed network capacity at all tandem switches, Multifunction Switches (MFSs), Softswitches (SSs), and critical dual-homed End Office (EO) switches and SCs.
4. The long-haul portion of the network infrastructure must be able to support a regional crisis in one theater, yet retain the surge capability to respond to a regional crisis occurring nearly simultaneously in another theater.
5. All F-F network infrastructure network connections supporting UC must have at a minimum, T1 bandwidth of 1.544 Mbps or greater.

6. All CE-R and/or AR interfaces in the direction of the CE-R support bandwidth metering and provisioning in accordance with the Four and Six Queue Bandwidth Provisioning Models as defined in [Tables 6.10-1](#) and [6.10-2](#).

NOTE 1: The Provisioning Model described below depicts how the AR routers are configured in the DISN. It is optional, but highly recommended, for MILDEPs to follow this bandwidth provisioning approach when provisioning network queues in support of UC services.

NOTE 2: Given the sensitivity and bandwidth constraints found in strategic and tactical environments, these types of B/P/C/S have the option to define their own provisioning model based on bandwidth availability and mission/operational recommendation needs.

**Table 6.10-1. DISN Four-Queue Bandwidth Provisioning Model**

AGGREGATE SERVICE CLASS	BANDWIDTH PROVISIONING
Network Control/Assured Real Time	25% Minimum Bandwidth
Assured Real-Time Video	15% Minimum Bandwidth
Preferred Elastic	40% Minimum Bandwidth
Best Effort	20% Minimum Bandwidth

**Table 6.10-2. DISN Six-Queue Bandwidth Provisioning Model**

AGGREGATE SERVICE CLASS	BANDWIDTH PROVISIONING
Network Control	5% Minimum Bandwidth
Assured Real Time	20% Minimum Bandwidth
Non-Assured Real Time	15% Minimum Bandwidth
Preferred Elastic	30% Minimum Bandwidth
Elastic	20% Minimum Bandwidth
Scavenger	10% Minimum Bandwidth

## 6.11 VOICE GRADE OF SERVICE

The Grade of Service (GOS) is defined in Appendix C, Definitions, Abbreviations and Acronyms, and References. In addition, the voice and video (UC only) GOS are calculated independently since the budgets associated with each are independent

1. The E2E network infrastructure must provide a GOS of P.00 (i.e., 0 sessions out of 100 will be “blocked” during the “busy hour”) for FLASH and FLASH OVERRIDE voice and video (UC only) sessions. This is also referred to as nonblocking service.
2. The E2E network infrastructure must provide a GOS of P.02 (i.e., 2 sessions out of 100 will be blocked during the busy hour) and P.01, respectively, during a 100 percent increase above

normal precedence usage for PRIORITY and IMMEDIATE voice and video (UC only) sessions at a minimum.

3. The E2E network infrastructure supporting UC must provide a peacetime theater GOS of P.07 (i.e., 7 voice sessions out of 100 will be blocked during the busy hour) or better, and an intertheater GOS of P.09 or better, as measured during normal business hours of the theaters for ROUTINE precedence voice and video (UC only) sessions traversing the network from an EO or SC EI and/or AS-SIP EI.
4. The CE Segment supporting UC must provide a GOS between the EO and any PBX users or between an SC and its subtended SC that do not exceed an additional blockage of P.02 for voice or video (UC video only) sessions.

## **6.12 TRAFFIC CONDITIONING CONSIDERATIONS**

### **6.12.1 Queuing Trust Considerations**

A trust boundary is the point within the network where DSCP markings begin to be accepted and packets are not dropped or overridden. The design objective is to enforce DSCP markings in accordance to the UCR as close to the endpoints as technically and administratively possible.

In order to achieve this objective, MILDEPs should adhere to the DSCP Plan as outlined in the UCR and described in section 6 of the UCR. MILDEP ASLAN infrastructure equipment should be configured to trust all UCR DSCP markings through every segment, or hop, of the network and must never attempt to markdown, reprioritize or drop UC DSCP markings. MILDEP ASLANs should not adopt COTS or vendor based QoS approaches, as these do not meet DoD requirements and guidance.

### **6.12.2 Queuing Policing, Scheduling & Markdown Considerations**

DISA GNSC enforces Committed Information Rates (CIR) for DISA WAN aggregation routers. MILDEPs should configure their devices and adhere to the GNSC CIRs and must take into account how the GNSC handles any surges above the CIR, when capacity is available.

The UCR recommends industry best practices when implementing congestion avoidance mechanisms. Any rate limiting, markdown or drop policy should support Expedited Forwarding (EF) with a strict priority above all other traffic classes for all inelastic real-time queues. It should also support Assured Forwarding (AF) with a Class-Based Weighted Fair Queuing (CBWFQ) approach. This provides for a certain level of delivery guarantee as long as the traffic does not exceed the CIR. Traffic that exceeds the CIR should face a higher probability of being rate limited or dropped if congestion occurs based on the priority and precedence of the traffic within that specific traffic queue. When coupled with Random Early Detection Queuing (RED) CBWFQ can:

1. Allows for lower level traffic flow fairness and prevents lower precedence queue bandwidth starvation by allowing for queue bandwidth reservation on converged networks
2. Lower precedence markings are only serviced when upper markings have been serviced
3. Aims to control the average queue size by indicating to the end hosts when they should slow down transmission of packets
4. Takes advantage of the congestion control mechanism of TCP by randomly dropping packets
5. When the device begins to sense periods of high congestion, the source starts to decrease its transmission rate. Assuming the packet source is using TCP, it will decrease its transmission rate until all the packets reach their destination, indicating that the congestion is cleared
6. Causes TCP to slow down transmission of packets. TCP not only pauses, but it also restarts quickly and adapts its transmission rate to the rate that the network can support
7. Distributes losses in time and maintains normal low queue depth while absorbing spikes
8. Drops packets when congestion occurs at a rate selected during configuration, when enabled on an interface.

UCR DSCP markings should never be remarked to a lower priority marking within the strategic network and on tactical networks, packets may be remarked as long as the appropriate markings are restored prior to entering the strategic network. All traffic should be configured in accordance with the Traffic Conditioning Specification (TCS) specified in the UCR Section 6.3.2.

## **6.13 UC NETWORK INFRASTRUCTURE SURVIVABILITY**

The following recommendations contribute to the survivability of the UC system:

No more than 15 percent of the B/P/C/Ss must be affected by any outage in the network. This includes issues such as overtaxing of processing capacity, link failure, and redundancy failover glitches.

## **6.14 VOICE SERVICE QUALITY**

1. Because intelligibility of voice communications is critical to C2, the voice service quality rating, on at least 95 percent of the voice sessions, will have an MOS IAW the following scenarios:
  - a. Fixed to Fixed Assured Voice – 4.0.
  - b. Fixed to Fixed Non-Assured Voice – 3.8.
  - c. Fixed to Deployable Assured Voice – 3.6.
  - d. Deployable to Deployable Assured Voice – 3.2.

2. The method used for obtaining the MOS must be IAW the DoD Information Technology Standards Registry (DISR), which currently aligns with the use of the E-Model and TSB-116-A (03/2006) for F-F scenarios, and P.862 (02/2001) for Deployable scenarios.